# BLIND COMPUTATION OF A SIGNAL USING SDR

M.BHUVANESHWARAN

BE-Electronics and Communication Engineering (III-year),
Knowledge Institute of Technology, Salem, Tamil Nadu, India,
bhuvaneshwaran1991@gmail.com

P.MANIGANDAN

BE-Electronics and Communication Engineering (III-year),
Knowledge Institute of Technology, Salem, Tamil Nadu, India,
manismart1991@gmail.com

R.SHANMUGASUNDARAM

Associate Professor, Department of ECE,
Knowledge Institute of Technology, Salem, Tamil Nadu, India,
rsece@kiot.ac.in.

**Abstract**

The most challenging aspect in signal processing is its hardware design. Today we are using multiple signals and each of it is at different frequency. The signal processing unit is a block in which the signal is processed to extract the exact message in the signal and it is mainly used to analyze each and every components of a signal. The block which is capable of doing all these things costs much. Thus the design of hardware components necessary to analyze the signal seems to be more difficult hence it needs huge amount of investment. In order to reduce the amount of money invested in the hardware design here in this paper we are presenting a new approach in signal processing called as software defined radio. Here the GNU RADIO acts as a software defined radio software and by using it we had successfully predicted the power, frequency spectrum and its peak amplitude of a speech signal.

**keywords :** SDR , GNU Radio , FFT.

## 1. Introduction

The analyses of unknown signals are becoming worse and it requires huge investment in hardware. The biggest problem is that if a particular hardware is designed for analyzing a particular set of frequencies then if the signal reaches beyond the limit then the hardware that is designed to analyze the signal won’t works better. Hence in these cases it is not capable of defining a particular signal with its exact measures. It might not be good if the signal is being processed further for an operation. Here arises the problem that, in order to analyze all the signals in a single system it is too complicate to design the hardware components.

Hence here approaches the idea of utilizing the personal computers in signal analysis and it results in the evolution of Software Defined Radio (SDR). The software defined radio is a radio that is built entirely or in large parts in software, which runs on a general purpose computer. It comprises of several blocks in it and all of it can be used on a received or on a predefined signal. It is merely 70% software based and 30% hardware based. Since it uses only 30% of hardware the amount invested in designing the hardware for signal processing unit is reduced. The SDR is capable of having the hardware components to be interfaced with it and it requires only the receiver block that is a simple receiver without having any processing blocks with it. All the processing blocks in the receiver can be defined manually using software. The software that incorporates the futures of SDR is GNU Radio.

## 2. GNU Radio

The software that incorporates the SDR is GNU Radio which is a open source software. It computes the given input signal by using different kinds of blocks which are present in it. It has more than 50 blocks in it to process the signal. It also offers special features that the user can also implement his own block in it. Thus it is very much useful in designing a particular block for the signal processing. The GNU Radio software has back end tool as Python and C++ Programming language. The Python programming language is a programming language which is similar to that of a C programming language. By using it we can create our own blocks by depending on the type of processing we need. The blocks which are in the GNU Radio are interconnected using python programming and all the blocks are coded using C++ software programming it is as shown in figure1.

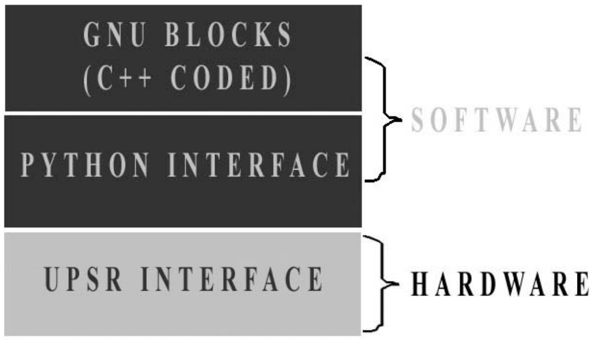


FIGURE 1: BASIC BLOCK DIAGRAM OF GNU RADIO.

As shown in the figure 1 the first two blocks GNU Blocks and the Python interface are software oriented that is it are all embedded within software. The hardware connected allows the user to externally apply input to the GNU radio. It generally uses UPSR module for transmitting and receiving the signal inside and outside of the GNU radio. It also has the signal source generator blocks which can be employed to generate the signal. Thus by generating the signal using signal source can be employed to test the same frequency signal to predict the amount of cost to be invested in processing the signal.

## 3. BLOCK DIAGRAM

Here we have assumed a basic signal processing unit which is used to predict the amount of power required to transmit the signal to another place. While transmitting the signal over a channel the noise that affects the signal is also added along the signal to predict the type of filter to be employed in the receiver to receive the signal in a better manner. Thus a small transmitting and receiving blocks are designed in the software itself and without investing the huge amount of money we can design all the components that are required to design the system it is as shown in figure 2.

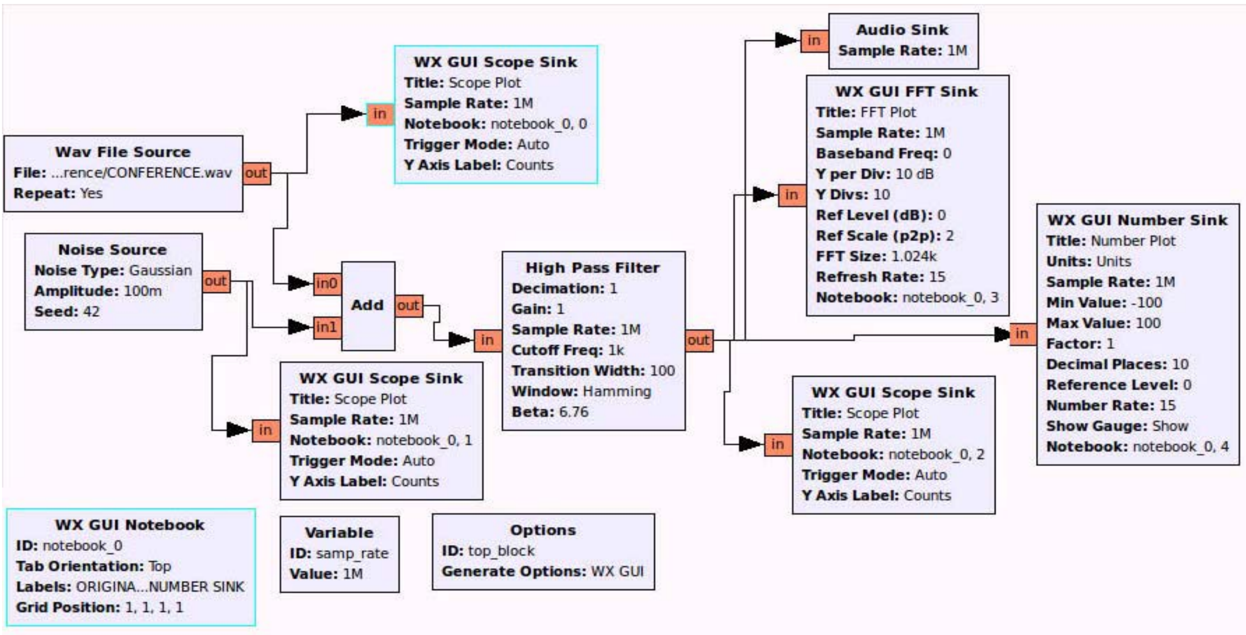


Figure 2: BLOCK DIAGRAM OF SIGNAL PROCESSING UNIT.

As shown in figure 2 the block diagram consists of two signal sources. The first one is the wave file source which is considered to give the input file in the format of wave file. Here we have considered a audio file source to analyze the amount of power needed to transmit it and also to determine the type of filter to be used to receive the signal clearly at the receiver. The second one is the noise signal source which is used to add noise signal to the input audio file. While transmitting the signal over a channel the signal gets affected by the noise, so in order to predetermine the type of filter to be used in the receiver the noise signal source is used. The adder block is used to add the noise signal into the audio signal which we have considered. The figure 3 shows the wave form of the input signal with noise and without noise.

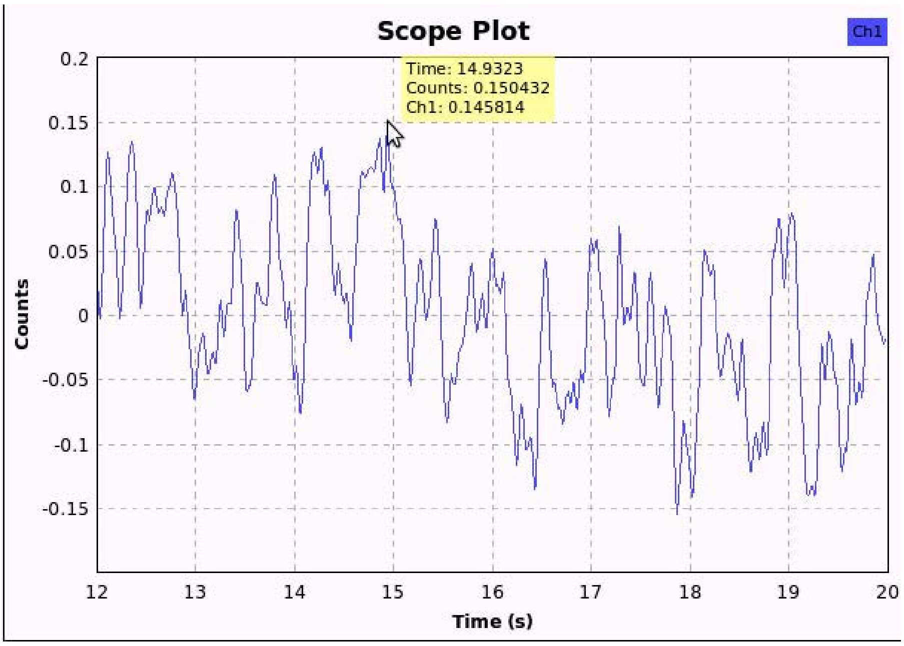


Figure 3.a: WAVE FORM OF ORIGINAL AUDIO SIGNAL CONSIDERED

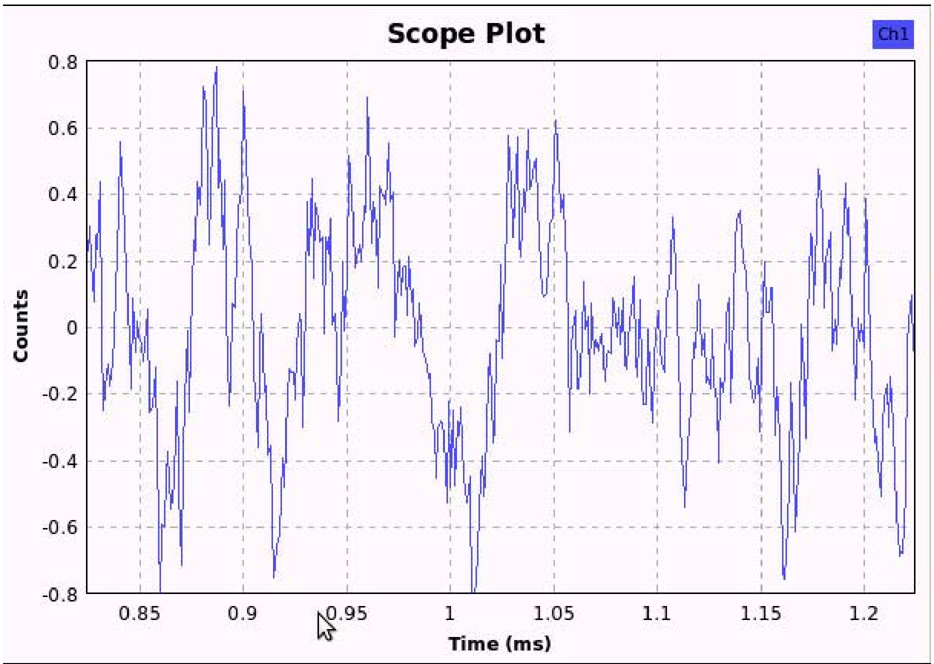


FIGURE 3.b: WAVE FORM OF ORIGINAL SIGNAL WITH NOISE

The figure 3.a shows the wave form of the original audio signal we have considered and from the figure it is evident that original signal has the maximum amplitude of 0.35V. After the addition of noise signal with the original signal the amplitude of the signal gets raised. In order to add the noise signal to the original signal the adder block of GNU Radio is used. The wave form of original signal after the addition of noise signal with it is shown in the figure 3.b. After the signal being received here we have applied the noise signal with the amplitude of about 0.1V hence the effect of noise signal on the original signal is of only a slight change. Hence we had used the high pass filter block to limit the noise signal to the considerable level. After that the signal received is viewed by various scope circuits to analyze the signal.

## 4. COMPUTATIONS:

### 4.1. FAST FOURIER TRANSFORM:

The Fast Fourier Transform is an excellent algorithm which is employed in almost all signal processing units. It offers several advantages over the conventional fourier transforms. The most commonly specified is the time consumption. The time taken by the FFT algorithm to analyze the signal is much lesser than that of the fourier transform algorithms. The internal characteristics of the signal are clearly examined in spectral form than in the time domain representations. Thus in order to compute several information of the signal the FFT algorithm is used as it consumes less time.

### 4.2. POWER:

The power of the signal plays a vital role in designing the medium through which the signal is being transmitted. For instance if the received signal needs to be transmitted to a particular distance hence the calculation of the power of the received signal plays a very important role. This is because without knowing the received signal power it is very difficult to predict the amount of voltage required to be amplified to transmit the same signal to the other end. Hence here we consider it to as one of the parameter to be analyzed.

Generally the power of the signal is determined by using the formula

$$\text{Power} = V^2/2$$

The V corresponds to the maximum amplitude of the signal. The maximum amplitude of the signal is determined from the scope of the signal.

## 5. RESULTS

Let us consider a sample audio signal which is being stored in the signal and its representation is shown in the figure 4.

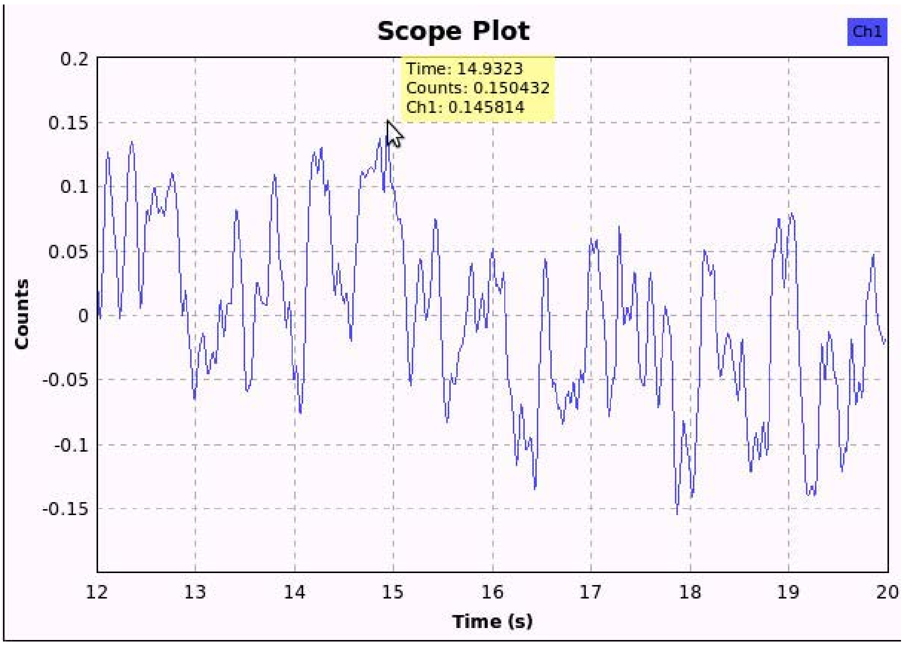


Figure 4- SCOPE OF THE INPUT AUDIO SIGNAL

The figure-4 shows the Scope spectrum of the given input audio signal from which the amplitude of the signal and its average variations can be easily predicted. The Scope Spectrum of the Noise Signal input is shown in the figure 5.

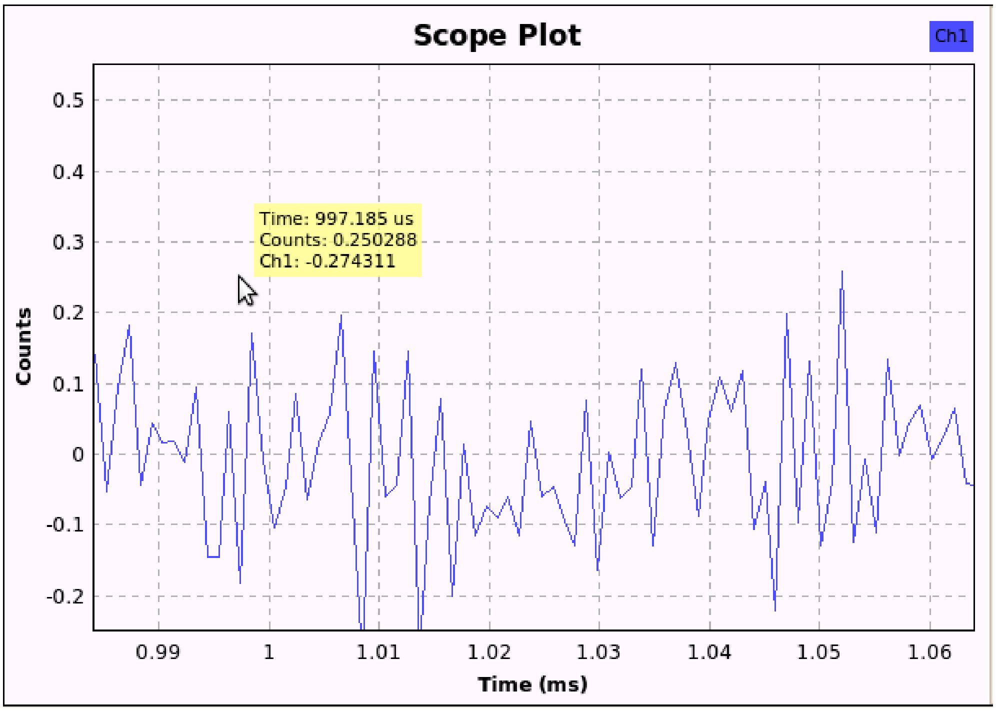


Figure-5: SCOPE SPECTRUM OF NOISE SIGNAL.

The FFT spectrum of the signal is shown in the Figure-6. The spectrum can be used to analyze the various internal characteristics of the signal.

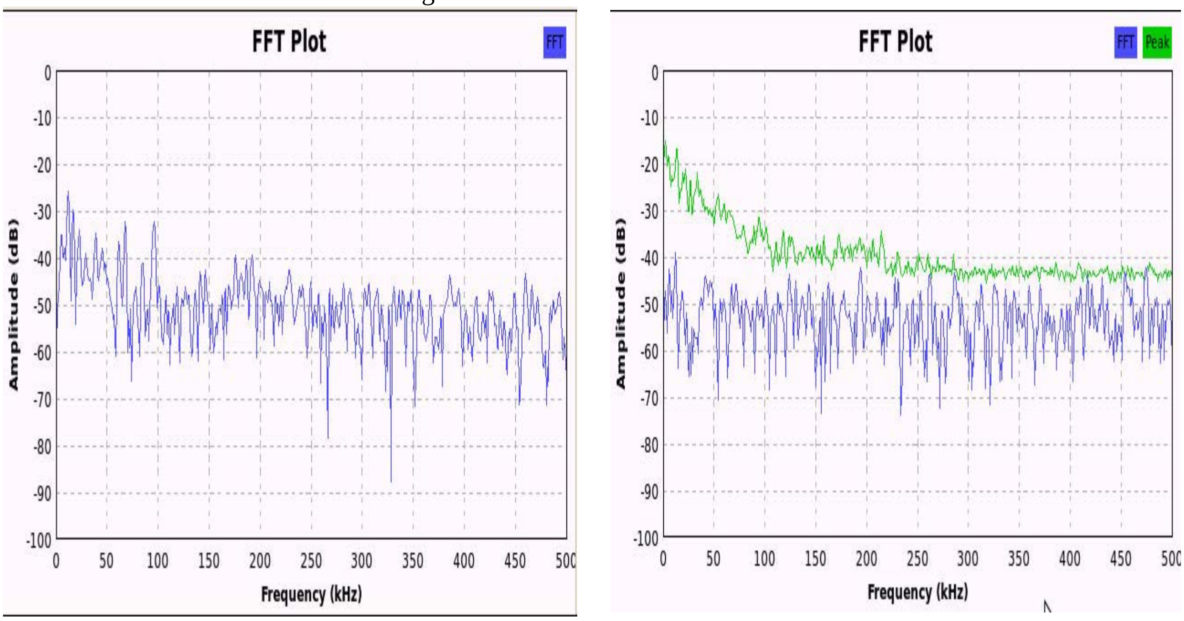


Figure-6.a: FFT Spectrum without Peak Value Hold Figure-6.b: FFT Spectrum with Peak Value Hold
Figure 6: FFT Spectrum of the Signal

The figure-7 shows the number plot of the signal in which the amplitude variations can be easily predicted in exact numbers by using this plot. In some cases it is very difficult to determine the exact value of the signal. Hence it is used. Two number plots are used to determine the amplitude of the signal at two stages that are one at the original signal stage and another one at the reception that is after passing through filter. The number plot at the receiver is considered because in certain cases the signal needs to be transmitted for further processing to some distance in that case the amplitude of the received signal is very important to determine the amount of power required to transmit it.

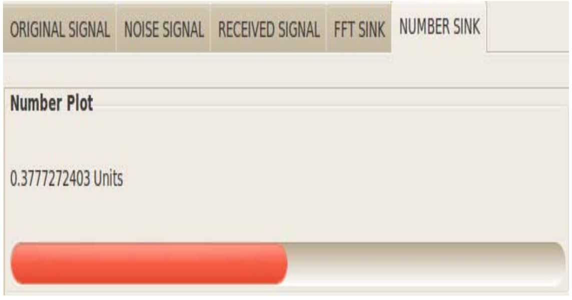


Figure 7.a: Number plot of the Received Signal

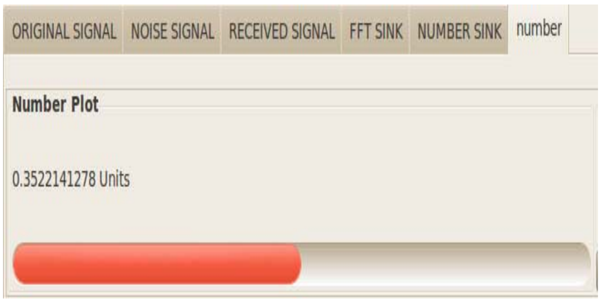


Figure 7.b: Number plot of the Original Signal

Figure-7 Number plot of the Signal.

And finally the power of the signal analyzed is calculated as

ORIGINAL SIGNAL POWER

Maximum Amplitude: 0.3522V

$$Power=V^2/2$$

$$Power = 0.3522^2/2 = 0.620 \text{ Watts.}$$

RECEIVED SIGNAL POWER

Maximum Amplitude: 0.3777V

$$Power =V^2/2$$

$$Power =0.3777^2/2 = 0.713 \text{ Watts.}$$

## CONCLUSION

As the analysis of a basic audio signal using SDR was successfully executed and the GNU-Radio software provides us lots and lots of facilities to work with it. In future works we planned to have the computation of the signal on the basics of modulation schemes used and to plot its PSR, SNR and exact measure of power needed.

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